# An Improved Audio-Frequency Bandpass Filter for Morse Code Reception

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In April of 2013 *ye scribe* proposed to Ed Wetherhold, W3NQN, that an unusual design be used for an audio-frequency bandpass filter for Morse Code ("CW") reception. The filter was to be added on to the output of receivers used for CW work. I saw a problem with many receivers used for CW and I also thought I had a solution to that problem. Ed had accumulated over the years a number of toroidal inductors with a nominal 77.2 mH value and was using these inductors for building audio-frequency bandpass filters. Hence his well-known *nom de plume* "Filter Builder" in the radio amateur fraternity. I suggested to him a passive filter design which used only one value of inductor. It was tailor-made to use some of Ed's stash of 77 mH inductors. Ed thought the design had merit and built several and put them out into the field for testing. Feedback revealed the concept to be quite satisfactory.

This paper looks into the filter's design rationale. It also explains why receivers with "rounded top" selectivity curves sound better to the CW operator, regardless of how that selectivity is achieved.

## The problem

The I.F. response of the receiver used by a typical CW operator has a flat top with a width of typically 400 Hz. Figure 1 illustrates the response of such an I.F. strip.



Figure 1 – Response of typical I.F strip

We have set the filter's center frequency to 500 Hz for these tests. The filter's bandwidth extends from 300 Hz to 700 Hz and drops off rather sharply beyond those limits. We are looking here at the response down at audio, not at the I.F. itself.

Now we will apply a single CW "dit", a burst, to that I.F. and see what exits. We can expect it to be distorted - and so to sound distorted - and in fact it appears as shown in Figure 2.

Transient Analysis



Figure 2 - A 500 Hz pulse 50 milliseconds long as it exits from that I.F.

Such a signal corresponds to a 24 WPM Morse signal, typical of the usual CW operator. The signal can be seen to be somewhat distorted at the leading and trailing edges. Time-domain analysis is being done here by *Elsie*, the filter design and analysis program. [1] *Elsie* also has another very useful option, that of revealing the *envelope* of that time-domain waveform. Using this option reveals the distortion more clearly. Figure 3 shows the envelope of this burst.

Transient Analysis



Figure 3 – The envelope of the burst shown in Figure 2.

The clearly-visible ringing seen in this graphic forecasts an audibly obnoxious sound, the kind of sound that the CW operator objects to.

This distortion is caused in large part by the group (envelope) delay distortion caused by the sharp cutoff at the I.F. filter band edges. This group delay distortion is a common byproduct seen in a filter with a sharp band edge. Group delay equalization can reduce these effects to only a limited extent.

Figure 4 shows the amplitude response of that I.F. filter along with the group delay.



Figure 4 – Magnitude response of the receiver's I.F. strip (upper plot) and the associated group delay (lower plot)

Consider the keyed CW signal as a modulated wave. As we go out from "carrier" (500 Hz in this discussion) the "sidebands" go down uniformly in amplitude. But if they are not handled correctly in both time and amplitude the signal will not sound right. If some of those sidebands happen to be delayed more than they should we will have ringing. This is what we see in Figures 2 and 3.

The ringing gets even worse as the CW speed increases to 30 or 35 WPM. If we can't change the shape of that filter's response than we'll have to do something else to improve the sound quality.

### A Solution

A way to improve the situation is to modify the *system* response by adding a second filter to the system – to the receiver output. This "add-on" filter would have a somewhat rounded top and it would be noticeably narrower than the usual I.F. bandwidth. The responses of this add-on filter are shown in Figure 5.





This add-on filter is somewhat narrower than the I.F. strip and has a more gentle response in both magnitude and time.

The system magnitude responses with and without the add-on filter are shown in Figure 6.



Figure 6 – The original receiver magnitude response (upper plot) and overall magnitude response with the add-on filter (lower plot)

The added audio bandpass filter adds some selectivity to the receiver. When we have added the new filter to the receiver we have narrowed the system bandwidth but – this is important - the system has a rounded top. This rounded top has an impact on signal components at band-edge as can be seen in Figure 7.



Figure 7 - Magnitude response of the overall system with the add-on filter (upper plot) and system group delay (lower plot)

That pair of plots show the system magnitude response (the upper plot) and group delay (the lower plot). The group delay of the sidebands at band edge is significant but notice that add-on filter has reduced the magnitude of the sideband components at bandedge to a very low amplitude. As a result, group delay problems at those frequencies are of much less consequence. We can expect the behavior of the system to be improved over the behavior of the receiver by itself.

Passing the burst through the system now results in an output as shown in Figure 8.



Figure 8 – The envelope of the burst when the add-on filter is used.

Compare this with Figure 3. The burst waveform shape is obviously improved. Reports from the units delivered to the field consistently report that a CW signal passing through this filter sounds better than does a signal from the receiver itself as might be expected.

The add-on filter effectively corrected the overshoot problem caused by the flat-topped I.F. filter in the receiver. This is because the add-on filter has removed the band edge components that would have caused the overshoots and it has done this in a controlled manner.

Another feedback item we see from the field is the reduction of noise when this filter is switched into the system. Let's see why this might be. When the filter's noise bandwidth is analyzed using *Elsie* we see that the noise bandwidth of the filter is somewhat under 200 Hz as seen here:

Noise Bandwidth Routine			
Starting frequency (Hz): 10	0	Lower frequency for 0.1% power:	281.2
Channing fragmann (Up)		Lower frequency for 1% power:	301.2
Stopping frequency (Hz). 90	0		
Number of steps:	k	Frequency of maximum transmission:	494
This network appears to be a bandpass or highpass Upper frequency for 1% pov		Upper frequency for 1% power:	678.79
type.		Upper frequency for 0.1% power:	704.39
I support Ellers should be swent station from a upper law from your			
Lowpass liners should be swept starting from a very low nequency. to above the 0.1% frequency.			
Bandhass fillers should be swent from below the lower 0.1% frequency			
to above the upper 0.1% frequency.			
Compute Noise Ba	ndwidth	Noise bandwidth: 181.81	
Return to Analysis page			
5			

Figure 9 – Noise bandwidth analysis

When the add-on filter's bandwidth determines the system bandwidth (is clearly narrower than the receiver's bandwidth) then it is largely responsible for the system noise bandwidth. The usual receiver's noise bandwidth is about 400 Hz, determined by the width if the I.F. bandpass filter.

Waveforms and Spectra

The pulsed waveform used for this paper has envelope rise and fall times of zero. The behavior of the system does not change until those times are increased to greater than about one millisecond. An interesting observation was made during the data-gathering portion of this paper in that a recommended rise and fall time of five milliseconds essentially negates the need for the filter such as the one under discussion. The next set of graphics is intended to illustrate this point. We are going to use *LTspice* [2] here for the analysis tool.

Figure 10 shows the shape of the burst with burst envelope rise and fall times set to zero. This is the signal as it is applied to the network being examined.



Figure 10 – Excitation applied to the add-on filter

This is the raw keyed CW signal as applied to the filter. The spectrum of this signal is shown in Figure 11.



Figure 11 – The spectrum of that raw (unshaped) keyed CW signal

The sidebands extend outward symmetrically and slowly drop off in amplitude. This is not at all a recommended situation. Key cllicks here we come.

Now we apply a five millisecond rise time and fall time specification to the envelope of the pulse. Figure 12 shows us what the resulting pulse looks like.



Figure 12 - The narrow banded pulse. The pulse was subjected to a five millisecond rise and fall time

This signal has a narrower bandwidth as can be seen in Figure 10.



Figure 13 – Spectrum of the pulse when shaped by the five millisecond rise and fall time constraint.

This signal occupies a quite noticeably narrower bandwidth. Passing this transmitted signal through our narrowband add-on filter should not result in significant alteration because the signal has already been made narrowband.

Compare this spectrum with that of Figure 11. All that has been done is to control the rise and fall times by using a single-pole lowpass (an R-C network) with a rise time to 90% of five milliseconds.

Now we'll apply that narrowband signal (shown in Figure 12) to our add-on bandpass filter.



Figure 14 – The narrowband pulse after it has passed through the add-on bandpass filter

Compare this waveform with that of Figure 12. The difference is small and we see little or no ringing, the item that forecasts poor sound for the CW operator.

The transmitted signal, when made narrowband by controlling its envelope, is hardly altered in wave shape when passed through the add-on filter. Bandwidth limiting of a transmitted CW signal is discussed in the ARRL Handbook. [3]

When the CW operator uses this add-on filter to listen to such a narrowband signal, the signal will be essentially unaltered except that other (interfering) signals will be significantly reduced in amplitude. In addition, noise will be reduced by using the add-on filter. Listening to signals which have fast rise and fall times will be made to sound better by virtue of "clicky" effects having been reduced in amplitude.

For these *LTspice* analyses of this narrowband situation we have not included the effects of the receiver's I.F. filter, which alters the wave shape only slightly.

## An Active Filter Equivalent

During the data-gathering portion of this paper, some correspondents asked about an active equivalent. Active bandpass filters are generally symmetrical on a geometric basis. Rephrased, they attenuate a given amount when the test frequency is changed by a given *factor* rather than by a specified frequency shift. As a simplistic example, the attenuation of the filter is typically the same when the test frequency is an octave below or an octave above the center frequency. When their responses are plotted on a linear frequency scale it can be seen that they commonly have poor attenuation on the high side of center.

An active filter design which behaves correctly, that is to say which has attenuation about the same on both sides of the center frequency, has its response shown in the dotted plot in Figure 15. The solid line in this graphic is the response of our passive filter and the dots show the response of the active filter (shown schematically in Figure 17).



#### Passive (solid line) and Active (dots) Comparison

Figure 15 – Responses of the passive version of the filter (solid line) and an active filter approximation (dotted) using a linear frequency scale

The width of the active version filter has been adjusted to be about the same as the passive filter we have been discussing and which has been fieldproven. The top is somewhat narrower, however, and is more rounded. This active version has not yet been subjected to field testing.

The output of this active filter when subjected to our test burst is shown in Figure 16.



Figure 16 – Output of the active filter with the applied burst

This filter handles the burst well and so it should sound quite nice. It is shown schematically in Figure 17.



Figure 17 – Schematic of an active equivalent of the filter

Note that we have two multiple-feedback bandpass filter sections and three lowpass filter sections. This complexity is required to achieve transmission as shown by the dotted plot in Figure 15. This active filter has a modest gain at its center frequency of 500 Hz. It is noticeably more complicated than the passive version and requires power although it is more compact.

The Passive Filter Schematic



Figure 18 – Schematic of the passive add-on bandpass filter

The above schematic is for the passive filter with a center frequency of about 500 Hz and shows optional input and output matching transformers.

Exact component values, computed to use inductors of 77.5 mH in value, are shown in this schematic. All of the inductors are of the same specified value; this was considered to be a very useful feature of the design. A real-world filter could use nearest 5% values; tighter tolerances would be preferred. This design is close to being arithmetically-symmetrical about the center frequency. Of possible interest to filter designers is that this trait holds true in this topology only for four-inductor designs.

The filter proper is to be driven from and terminated in impedances of about 125 ohms. Transformers are shown on the input and output for matching from an 8-ohm source and to an 8-ohm load. If such a transformer is used on the input for impedance-matching, be aware that distortion generated by the transformer can cause false responses to appear. If driven into some degree of nonlinearity, the transformer can act as a frequency multiplier and the following filter can then pick off an odd harmonic. In such a case, signals at one-third of the design center frequency can show up in the output. This can hold true for other odd harmonics (fifth, for example) as well.

There are three things working in concert for this passive design that make it sound good to the CW operator. Each, on its own, makes only a modest contribution to the final result.

- The design topology is mesh-capacitor-coupled; all inductors are of the same value. It happens to be arithmetically-symmetrical: the "sidebands" are treated equally on each side of "carrier."
- The design family is Butterworth. This is a "gentle" family and has a smooth top in the passband and a smooth descent into the stopband.
- The inductor Q values are somewhat low, about 40. This, too, contributes to a smooth, rounded passband (the "top") and a smooth descent into the stopband.

Under many - perhaps most - circumstances those attributes may be of dubious importance. In this case, however, they work together and so we have a nice filter with numerous favorable reviews on eham.net [4].

Ed Wetherhold has offered to supply components for this filter. [5]

[1] Elsie – A filter design and network analysis program.

The quite-capable free Student Edition is included on the CD accompanying the ARRL Handbook and is included as part of the downloadable package available on the ARRL website:

http://www.arrl.org/arrl-handbook-reference

The full Professional Edition of Elsie is available from Tonne Software: <u>http://tonnesoftware.com/elsie.html</u>

[2] LTspice – A general-purpose nodal circuit analysis tool available from Linear Technology Corp.: http://www.linear.com/designtools/software/#LTspice

[3] ARRL Handbook - Digital Modulation; see Fig. 8.11 and associated text

- [4] <u>http://www.eham.net/reviews/detail/58</u>
- [5] ed.w3nqn@comcast.net

James L. Tonne holds the Extra class callsign W4ENE, and was first licensed in 1951. His current amateur radio interests are largely focused on speech processing and filter design. He has written several articles for QST and QEX and was a major contributor to the RF and Filters chapter in the ARRL Handbook. He is the author of the Tonne Software package on the CD accompanying the ARRL Handbook and included as part of the downloadable package available on the ARRL website.